



Media Gateways

- SS7 ISUP (including several national variants)
- ☑ Up to 16 SS7 Signalling links
- SIP RFC3261
- ☑ H.248 (Megaco)
- SIGTRAN M2UA
- **☑** H.323

Supports a wide range of VoIP and wireless codecs

- G.711 G.723.1
- G.726 iLBC
- G.729AB GSM
- G.722 AMR
- G.722.1

Sophisticated call routing via XML scripts

Management

- Web GUI
- CLI
- Statistics/logs
- PCAP Protocol Capture for analysis via Wireshark
- RADIUS Protocol
- SNMP for Layer 1 reporting
- ☑ Distributed architecture and signalling relay
- ☑ Convenient binary package or ISO image

Sangoma's NetBorder SS7 to VoIP Gateway software provides full-featured, carrier-class VoIP deployments while leveraging the flexibility of standard computing platforms and operating systems.

SS7 TO VOIP MEDIA GATEWAY SOFTWARE

The NetBorder SS7 to VoIP Gateway allows telecom service providers to introduce VoIP in their networks in the most cost-effective and flexible way. This is simply accomplished by combining the software with Sangoma's award-winning digital E1/T1 and transcoding boards on standard computing servers. The combination works as a full-fledged SS7 to VoIP gateway, with the flexibility and expandability of software.

The solution supports up to 32 E1/T1 per server. For installations up to 256 E1/T1, distribution across multiple servers provide maximum flexibility to support growth.



BENEFITS

- Wide range and support of SS7 PSTN protocols and variants
- Scalable
- Flexibility of software deployments instead of monolithic hardware platforms
- Low cost installation leveraging Open Source and off-the-shelf components
- Robust implementation with distribution, failover and redundancy

CONTINUE READING »



TECHNICAL SPECIFICATIONS

PSTN Protocols:

- SS7-ISUP
 - » ITU, ANSI, Bellcore, UK, China, India, France, and Russian variants
- Up to 16 A or F signalling links
- Up to 16 Originating Point Codes
- Up to 16 Destination Point Codes
- Up to 16 Linksets
- ISUP relay for larger configurations

Network Interfaces via Sangoma Telephony Hardware Up to 32 E1/T1 (960 ports) per server via digital hardware:

- A101D / A101DE 1-port E1/T1
- A102D / A102DE 2-port E1/T1
- A104D / A104DE 4-port E1/T1
- A108D / A108DE 8-port E1/T1

Extend capacity over 960 ports and single server via ISUP relay feature. Sangoma recommends hardware echo cancellation option.

VoIP Protocols:

- SIP V2/RFC3261 Megaco/H248
- SIGTRAN M2UA RFC 3331 H.323
- SCTP RFC 2960

Call Routing:

• Configurable and extendable XML-based dial plan and routing rules

Operating System Support:

• 32 bit and 64 bit Linux; CentOS recommended

NetBorder

• Software delivered as binary package or ISO image

Minimum Server Requirements:

- Varies with size of deployment
- Dual Core CPU with 2GB or RAM
- Consult Sangoma Sales for specifics

Media Processing:

Transcoding:

• Wide range of codecs support via Sangoma transcoding hardware (D100/D150/D500):

»	G.711	» G.723.1
»	G.711	» G.723.2

- » G.726 » iLBC
- » G.729AB » GSM
- » G.722 » AMR
- » G.722.1
- T.38 Fax Relay

Echo Cancellation:

• G.168-2002 with 128ms tail

DTMF Detection and Generation:

- RFC2833 Tone relay
- In-band
- DTMF detection and generation

Management and Configuration:

- Web GUI
- Command line interface
- Call detail records in XML format
- Detailed logs with configurable filesize & autorotation
- SNMP
- Radius

Support and Professional Services:

Whether you need technical support and software maintenance, training, consultation and installation services, Sangoma can help you. Contact your Sales representative for more information.Sales representative for more information.



Distributed Architecture for Large Scale Deployments